

Award #1 Title
**Acoustic Communications 2011 Experiment: Deployment Support and Post
Experiment Data Handling and Analysis**

Award #2 Title
**Exploiting Structured Dependencies in the Design of Adaptive Algorithms for
Underwater Communication**

Award #3 Title
**Coupled Research in Ocean Acoustics and Signal Processing for the Next
Generation of Underwater Acoustic Communication Systems**

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LONG-TERM GOALS

A high performance, versatile, and reliable underwater communications capability is of strategic importance to the U.S. Navy. Operational scenarios involving the use, monitoring, and coordination of multiple undersea assets, both manned and unmanned and mobile and fixed, are significantly enhanced by the ability to communicate quickly, reliably, in a wide range of environmental conditions, and with minimal constraints on the actions of the platforms involved. The long-term goal of the effort in underwater acoustic communications is to integrate research in the areas of physical oceanography, ocean acoustics, signal processing, and communications theory to

1. develop a underwater communications capability that can be deployed on and used by a wide range of platforms with minimal required external infrastructure and which achieves reliable and high performance under a wide range of environmental and operational conditions, and
2. develop an communications performance prediction capability that enables commanders to

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reliably predict the performance of deployed communications systems as a function of environmental conditions and the location, velocity, and capabilities of the deployed assets.

OBJECTIVES

The objectives of the work under Grant #N00014-111-0426 are to support the Long Term Goals in the context of underwater **acoustic** communications systems. The objectives include

1. Work with Dr. William Hodgkiss, Scripps Institution of Oceanography, to conduct the KAM11 field experiment and gather a significant data set of spatially and temporarily coincident environmental and acoustic data with the acoustic data including receptions of ambient noise, general purpose channel probe signals, and specialized communications signals that are to be evaluated for specific applications. Perform appropriate quality control not the data and make it available to KAM11 participants.
2. Develop and validate models of the impact of sea surface and upper ocean boundary layer processes on the performance of underwater acoustic communications systems.
3. Develop methods of detecting and synchronizing to known underwater acoustic communications signals at very low signal to noise ratios.
4. Develop, analyze and predict the performance of adaptive decision feedback equalization algorithms that are suitable for use with receive arrays with large numbers of elements and which offer improved performance and reduced computational complexity when compared to standard full rank, hard decision directed adaptive equalization algorithms. Develop new methods for predicting the performance of adaptive equalization algorithms in the underwater acoustic environment and use them to determine optimal equalizer and array configurations in realistic environments.

APPROACH

The approach taken to consists of a combination of analysis of field data, the development of new theory for adaptive signal processing algorithms, and the development, testing and analysis of new algorithms. The specifics are described below with paragraph numbers corresponding to the paragraph numbers in the preceding section (OBJECTIVES).

1. The conducting of the KAM11 field experiment was a straight forward exercise in experiment planning and execution with a large group of PIs. The approach was to gather PI requirements signal transmissions and signal and environmental measurements, coordination with the ship and PMRF, working with technicians at WHOI and Scripps to plan deployment and recovery operations, and scheduling of the min-experiments that we incorporated into the overall KAM11 experiment.

Post-cruise, the approach was to analyze the data in the sequence of transmissions, identify time periods when there were either transmission or reception problems, correct the problems when possible, review the environmental data to determine a representative set of time periods, and then quality check the data in those time periods and make it available to other KAM11 experiment participants.

2. Objective 2 is being pursued in collaboration with Dr. Grant Deane, Scripps Institution of Oceanography and Dr. Andone Lavery, Woods Hole Oceanography. The approach is to develop analytical and numerical models to characterize the channel impulse response from a transmitter to a receiver resulting from surface scattered arrivals. This characterization includes intensity fluctuations due to surface shape and the path length fluctuations due to surface motion. Also included are the effects of near surface bubbles on the intensity and time-variability of the surface scattered arrivals. This characterization is carried out with the analysis of derived expressions, numerical evaluation of expressions, and numerical simulations. The derived expressions and numerical models are validated by comparison between predicted impulse responses and those measured during the KAM11 and SPACE08 field experiments. Environmental measurements taken during those two experiment is used as the inputs to the numerical simulations used to generate the impulse response predictions. This work is pursued with the support from award 1.
3. The approach taken to achieving low-snr detection of and synchronization to communications signals it to hypothesize that a specific location (time) in a received signal is the starting point for a known detection and synchronization portion of a received signal and train a specially constructed adaptive equalizer to demodulate the signal known signal. If the signal is actually present and it is a bpsk modulated signal, we would expect that the equalizer would achieve a bit error rate of something less than 0.5 (random guessing for a bpsk signal) indicating the detection of and synchronization to a communication signal. If there is no communications signal present, then the equalizer would be expected to achieve a bit error rate of approximately 0.5. indicating that there has been no detection. This work is pursued with the support from award 1.
4. Adaptive, multi-channel, phase-coherent equalizers reliably achieve higher performance in terms of achievable data rates in underwater acoustic channels than other forms of signal detection and demodulation algorithms. However, in their customary form they have a large number of parameter that need to be adjusted. This large number of parameters results in both a high computational complexity of the equalizer adaptation algorithm and the need to utilize a long averaging interval in order to achieve stable operation. This long averaging interval decreases the rate of unmodeled acoustic channel fluctuations that can be successfully tracked and compensated for by the equalizer. Finally, the use of such equalizer in turbo-equalization algorithms posses great promise for future performance gains. Yet, in the underwater acoustic environment where the channel can change significantly over the duration of a communications packet. Thus, the equalizer must be adapted in each iteration of the turbo algorithm during which it has imperfect knowledge of the transmitted data symbols which are needed to guide the adaptation algorithm.

The approach to addressing the "large number of parameters" problem will be to develop beamspace processing methods and adaptive subarray processing methods for transforming the single high dimensional adaptation problem into one or more lower dimensional problems. Supporting results in the field of *Random Matrix Theory* will be developed to assist in the guidance of optimal methods and the prediction and analysis of their performance. The approach to developing an adaptive equalizer capable of operating within the iterative framework of a turbo-equalizer will be to use soft symbol information in the equalizer adaptation process place more weight on symbol decisions which are believed to be reliable and to discount the decisions which are not. The Expectation-Maximize framework will serve as a starting point for this development. This work will be pursued primarily with the support from Awards 1 and 3. Finally, the covariance matrix of the filter regressor vectors (the frequency domain input signals to both

the feedforward and feedback filters) posses a predictably sparse structure that will be exploited to improve performance and/or reduce computational complexity with respect to an adaptive equalizer that does not exploit any particular structure of the matrix. This later approach will be pursued primarily with the support of Award 2.

WORK COMPLETED

This year the completed work has been in three areas. The first are concerns the use of Random Matrix theory to characterize the performance of adaptive processing algorithms which use sample correlation matrixes in the adaptation of algorithm parameters. We completed work including characterizing the performance of multichannel linear and decision feedback equalizers as a function of filter parameters (the number of taps in each of the filters), array configuration (spacing between elements in a linear array), and the acoustic propagation conditions. This work has started to be extended under the support of contract #N00014-14C-0230 to create a model of the sample covariance matrix of the input signal that both explicitly models the time-variability of the acoustic channel and is amenable to analysis using the tools of asymptotic random matrix theory. A new model has been created but has yet to be analyzed and evaluated. The second area in which work as been completed concerns the exploitation of the predictable sparse structure of the input signal covariance matrix in a frequency domain equalizer.

RESULTS

The work on random matrix theory has let to the interesting result that the optimal configuration of both linear and decision feedback multi-channel equalizers in a time-varying ocean environment utilized filter lengths that are much shorter than the delay spread of the acoustic channel. Thus, it appears that the goal of canceling the majority of the intersymbol interference is accomplished primarily by the spatial processing made possible by reception of the signal at an array of hydrophones rather than at a single hydrophone. This work was presented at the UComms 2014 conference by Milutin Pajovic and myself.

The work on the exploitation of sparse structured covariance matrices in equalizer adaptation yielded significant results in several areas. It was first demonstrated the that the sparse structure exhibited in the covariance matrix of the frequency domain inputs to the feedforward and feedback filters is also present in the inverse of this covariance matrix. This implies that the sparse structure influences the solution to the least squares adaptation problem. Second, Recursive Least Squares (RLS) algorithms which exploit this structure were derived and analyzed. The first stage of this work was presented at the ICASSP 2014 conference by Atulya Yellepeddi and myself.

IMPACT/APPLICATIONS

All of the above results are directed towards improving the performance of underwater acoustic communications systems. In particular, these results will allow future systems to operate reliably at lower signal to noise ratios and achieve a higher level of reliability and data transmission rate in challenging environments than is possible with current systems. The RMT work and work on surface scattering will facilitate a theory based approach to optimal spacing and partitioning of arrays and the two stage processing of subarrays of sensors and the combining of their outputs. The work on sparse structure covariance matrices in adaptive equalization will improve the performance of adaptive

equalizers in challenging environments such as those with low SNRs or rapid environmental fluctuations.

RELATED PROJECTS

The work under this grant is closely related to the MURI funded project titled, "Underwater Acoustic Propagation and Communications: A Coupled Research Program", ONR Grant Number N00014-07-10738. Funds from both grants were used to pay for the KAM11 experiment and the post-experiment processing and analysis of the data as well as the development and analysis of channel equalization algorithms.

PUBLICATIONS

A. Yellepeddi, J. Preisig, "Adaptive Equalization in a Turbo Loop", submitted to *IEEE Trans. on Wireless Communication*, revised after first review.

A. Yellepeddi, J. Preisig, "Approximate least squares parameter estimation with structured observations", in *Proc. IEEE Int. Conf. on Acoustics, Speech, and Signal Processing*. Florence, Italy, May 4 - 9, 2013, pp. 5671 - 5675.

M. Pajovic, J. Preisig, "Optimal Multi-Channel Equalizer Design for Underwater Acoustic Communications", to appear in *Proc. of 2nd Underwater Communications Conference*, Sestri Levante, Italy, September 3 - 5, 2014.